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S/N 10/044210

PATENTIN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant:	Rademacher, et al	Examiner:	To be assigned
Serial No.:	10/044210	Group Art Unit:	To be assigned
Filed:	12/13/01	Docket No.:	635.347US01
Title:	METHOD AND APPARATUS FOR REDUCING RANDOMLY, CONTINUOUS, NON-STATIONARY DISTORTIONS IN AUDIO SIGNALS		

CERTIFICATE OF MAILING UNDER 37 C.F.R. §1.8

I hereby certify that this correspondence is being deposited with the United States Postal Service with sufficient postage as first class mail in an envelope addressed to: Assistant Commissioner for Patents, Washington, DC 20231 on April 17, 2002.

Michael B. Lasky

Name

Signature

PRELIMINARY AMENDMENT

Box Missing Parts
Assistant Commissioner for Patents
Washington, D.C. 20231



Dear Sir:

Please enter the following preliminary amendment into the above-referenced application.

SPECIFICATION

An error was noted in the title of the application with regard to the spelling of "continuous". Please replace [continous] with the correct spelling of continuous.

CLAIMS

Please amend claims 1-16 as follows. A clean copy of the amended claims is included below. A marked up copy of the entire claim set is included in Appendix A.

1. (Amended) A method of reducing random, continuous, non-stationary noise in audio signals which are present in discrete form or which are obtained from the sampling of an analog audio signal with random, continuous, non-stationary noise,

wherein the noisy audio signal is filtered by means of a filter function wherein the filter function is determined dynamically having regard to the current properties of the noisy audio signal and/or its constituent parts, and that the filter function is limited dynamically having regard to the current properties of the noise component contained in the noisy audio signal.

2. (Amended) A method as set forth in claim 1 wherein an estimate of the noise component of the noisy audio signal is produced, which describes the change in respect of time of the noise, the unrestricted filter function $H_G(m, l)$ is determined in per se known manner from the estimate of the noise component, a restriction function $\gamma_{SF}(m, l)$ is produced in dependence on the estimated noise component of the noisy audio signal, and a restricted filter function H_b is produced in accordance with the following:

$$H_b = H_G^{dyn}(m, l, \gamma_{SF}(m, l)) = \begin{cases} H_G^{dyn}(m, l) & \text{for } H_G^{dyn}(m, l) > \gamma_{SF}(m, l) \\ \gamma_{SF}(m, l) & \text{other} \end{cases}$$

and is used for filtering the noisy audio signal,

wherein m is the considered discrete spectral frequency or another parameter which permits an equivalent representation of the signals and l is the discrete time of the respectively considered signal block in the case of block-wise signal processing, wherein a block may also include only one sample value.

3. (Amended) A method as set forth in claim 1 wherein the restriction function $\gamma_{SF}(m, l)$ is produced in dependence in respect of time on the estimate which is variable in respect of time of the noise component of the noisy audio signal.

4. (Amended) A method as set forth in claim 3 wherein the restriction function $\gamma_{SF}(m, l)$ is produced in dependence in respect of time on the instantaneous noise power which is variable in respect of time of the estimated noise component of the noisy audio signal.

5. (Amended) A method as set forth in claim 1 wherein the restricted filter function is produced in one method step.

6. (Amended) A method as set forth in claim 1 wherein filtering of the noisy audio signal is executed in the time domain, in the frequency domain or in another mathematically describable signal space.

7. (Amended) A method as set forth in claim 1 wherein the unrestricted filter function $H_G^{dyn}(m, l)$ is determined in accordance with an approach according to Wiener,

in which the mean quadratic error between useful signal and estimate is used as the approximation criterion.

8. (Amended) A method as set forth claim 1 wherein the unrestricted filter function $H_G^{dyn}(m, l)$ is determined in accordance with the amplitude subtraction method.

9. (Amended) A method as set forth claim 1 wherein the noisy audio signal $x(k)$ is transformed into the frequency domain, then the noise component $N(m, l)$ of the transformed noisy audio signal $X(m, l)$ is estimated, the unrestricted filter function $H_G^{dyn}(m, l)$ and the restriction function $\gamma_{SF}(m, l)$ is produced and the restricted filter function H_b is formed therefrom, then the transformed noisy audio signal $X(m, l)$ is multiplied by the restricted filter function H_b and then transformed back into the time domain.

10. (Amended) A method as set forth in claim 1 wherein the filter function $H_G^{dyn}(m, l)$ is determined by means of a known approach utilizing an estimate $\hat{\Phi}_{NN}(m, l)$ of the instantaneous auto-noise power density.

11. (Amended) A method as set forth in claim 10 wherein the estimate $\hat{\Phi}_{NN}(m, l)$ of the instantaneous auto-noise power density is determined from a weighting of the estimate $\hat{\Phi}_{NN}(m)$ with a time-dependent weighting factor $\alpha(m, l)$ to give:

$$\hat{\Phi}_{NN}(m, l) = \alpha(m, l) \cdot \hat{\Phi}_{NN}(m).$$

12. (Amended) A method as set forth in claim 11 wherein the weighting factor $\alpha(m, l)$ is ascertained in accordance with:

$$\alpha(m, l) = \frac{\min(|X(m, l)|^2)}{\min(\hat{\Phi}_{NN}(m))}$$

wherein $X(m, l)$ is a representation of the noisy audio signal.

13. (Amended) A method as set forth in claim wherein the dynamic restriction function $\gamma_{SF}(m, l)$ is determined as:

$$\gamma_{SF}(m, l) \sim (\alpha(m, l))^\beta, \text{ with } -5 < \beta < 5.$$

14. (Amended) A method as set forth in claim 13 wherein $\beta = 1/2$.

15. (Amended) Apparatus for reducing random, continuous, non-stationary noise in audio signals which are present in discrete form or which are obtained from the sampling of an analog audio signal with random, continuous, non-stationary noise,

wherein the noisy audio signal is filtered by means of a filter function whereby a device (4; 22) for estimating the noise component of the noisy audio signal, wherein said estimate takes account of the change in respect of time of the statistical properties of the noise,

a device (8; 30) for producing an unrestricted filter function H_G^{dyn} in per se known manner having regard to the previously ascertained estimate of the noise component which takes account of the changes in respect of time of the statistical properties of the noise,

a device (24, 40) for producing a time-dependent restriction function γ_{SF} in dependence on the estimated noise component of the noisy audio signal, and

a device (7; 40) which produces a restricted filter function H_b from the unrestricted filter function H_G^{dyn} and the time-dependent restriction function γ_{SF} , and

a filter (7; 50) which filters the noisy audio signal with the restricted filter function H_b .

16. (Amended) Apparatus as set forth in claim 15 wherein the device (9; 40) produces the restricted filter function H_b in accordance with:

$$H_b = H_G^{dyn}(m, l, \gamma_{SF}(m, l)) = \begin{cases} H_G^{dyn}(m, l) & \text{for } H_G^{dyn}(m, l) > \gamma_{SF}(m, l) \\ \gamma_{SF}(m, l) & \text{other.} \end{cases}$$

REMARKS

The above preliminary amendment is made correct a spelling mistake in the title of the application and to remove multiple dependencies from claims.

Applicant respectfully requests that this preliminary amendment be entered into the record prior to calculation of the filing fee and prior to examination and consideration of the above-identified application.

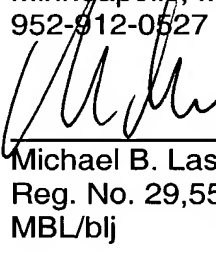
If a telephone conference would be helpful in resolving any issues concerning this communication, please contact Applicant's attorney of record, Michael B. Lasky at 952-253-4100.

Respectfully submitted,

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Date: April 17, 2002

By:



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Appendix A

Marked Up Version of Entire Claim Set

1. (Amended) A method of reducing random, continuous, non-stationary noise in audio signals which are present in discrete form or which are obtained from the sampling of an analog audio signal with random, continuous, non-stationary noise, wherein the noisy audio signal is filtered by means of a filter function, [characterised in that] wherein the filter function is determined dynamically having regard to the current properties of the noisy audio signal and/or its constituent parts, and that the filter function is limited dynamically having regard to the current properties of the noise component contained in the noisy audio signal.

2. (Amended) A method as set forth in claim 1 [characterised in that] wherein an estimate of the noise component of the noisy audio signal is produced, which describes the change in respect of time of the noise,
the unrestricted filter function $H_G(m, l)$ is determined in per se known manner from the estimate of the noise component,
a restriction function $\gamma_{SF}(m, l)$ is produced in dependence on the estimated noise component of the noisy audio signal, and a restricted filter function H_b is produced in accordance with the following:

$$H_b = H_G^{dyn}(m, l, \gamma_{SF}(m, l)) = \begin{cases} H_G^{dyn}(m, l) & \text{for } H_G^{dyn}(m, l) > \gamma_{SF}(m, l) \\ \gamma_{SF}(m, l) & \text{other} \end{cases}$$

and is used for filtering the noisy audio signal,

wherein m is the considered discrete spectral frequency or another parameter which permits an equivalent representation of the signals and l is the discrete time of the respectively considered signal block in the case of block-wise signal processing, wherein a block may also include only one sample value.

3. (Amended) A method as set forth in [one of claims 1 and 2 characterised in that] claim 1 wherein the restriction function $\gamma_{SF}(m, l)$ is produced in dependence in respect of time on the estimate which is variable in respect of time of the noise component of the noisy audio signal.

4. (Amended) A method as set forth in claim 3 [characterised in that] wherein the restriction function $\gamma_{SF}(m, l)$ is produced in dependence in respect of time on the instantaneous noise power which is variable in respect of time of the estimated noise component of the noisy audio signal.

5. (Amended) A method as set forth in [one of claims 1 through 4 characterised in that] claim 1 wherein the restricted filter function is produced in one method step.

6. (Amended) A method as set forth in [one of the preceding claims characterised in that] claim 1 wherein filtering of the noisy audio signal is executed in the time domain, in the frequency domain or in another mathematically describable signal space.

7. (Amended) A method as set forth in [one of the preceding claims characterised in that] claim 1 wherein the unrestricted filter function $H_G^{dyn}(m,l)$ is determined in accordance with an approach according to Wiener, in which the mean quadratic error between useful signal and estimate is used as the approximation criterion.

8. (Amended) A method as set forth in [one of the preceding claims characterised in that] claim 1 wherein the unrestricted filter function $H_G^{dyn}(m,l)$ is determined in accordance with the amplitude subtraction method.

9. (Amended) A method as set forth in [one of claims 1 through 8 characterised in that] claim 1 wherein the noisy audio signal $x(k)$ is transformed into the frequency domain, then the noise component $N(m,l)$ of the transformed noisy audio signal $X(m,l)$ is estimated, the unrestricted filter function $H_G^{dyn}(m,l)$ and the restriction function $\gamma_{sf}(m,l)$ is produced and the restricted filter function H_b is formed therefrom, then the transformed noisy audio signal $X(m,l)$ is multiplied by the restricted filter function H_b and then transformed back into the time domain.

10. (Amended) A method as set forth in claim 1 [characterised in that] wherein the filter function $H_G^{dyn}(m,l)$ is determined by means of a known approach [utilising] utilizing an estimate $\hat{\Phi}_{NN}(m,l)$ of the instantaneous auto-noise power density.

11. (Amended) A method as set forth in claim 10 [characterised in that] wherein the estimate $\hat{\Phi}_{NN}(m,l)$ of the instantaneous auto-noise power density is determined from a weighting of the estimate $\hat{\Phi}_{NN}(m)$ with a time-dependent weighting factor $\alpha(m,l)$ to give:

$$\hat{\Phi}_{NN}(m,l) = \alpha(m,l) \cdot \hat{\Phi}_{NN}(m).$$

12. (Amended) A method as set forth in claim 11 [characterised in that] wherein the weighting factor $\alpha(m,l)$ is ascertained in accordance with:

$$\alpha(m,l) = \frac{\min(|X(m,l)|^2)}{\min(\hat{\Phi}_{NN}(m))}$$

wherein $X(m,l)$ is a representation of the noisy audio signal.

$$H_b = H_G^{dyn}(m, l, \gamma_{SF}(m, l)) = \begin{cases} H_G^{dyn}(m, l) & \text{for } H_G^{dyn}(m, l) > \gamma_{SF}(m, l) \\ \gamma_{SF}(m, l) & \text{other.} \end{cases}$$